BBC RD 1978/5

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REPORT

An improved correlation audio combiner for use with H.F. diversity receivers

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THE BRITISH BROADCASTING CORPORATION March 1978

BBC RD 1978/5
UDC 621.391.85:
621.391.812.31

AN IMPROVED CORRELATION AUDIO COMBINER FOR USE WITH H.F. DIVERSITY RECEIVERS

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Summary

This Report describes improvements that have been made to a correlation audio combiner that was described in an earlier Report.

Subjective tests are described which demonstrate that, under conditions of frequency-selective fading, the performance of the improved combiner is superior to that of present combining methods.

Issued under the authority of

GDMonteath

Head of Research Department

Research Department, Engineering Division,
BRITISH BROADCASTING CORPORATION

March 1978

(EL-137)

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Section	Title	Page
	Summary	Title Page
1.	Introduction	1
2.	Correlation audio combiner	1
	2.1. Correlation and combining	1
	2.2. Gain-compensation	3
	2.3. Master signal selection	3
3.	Use of receiver A.G.C.	3
4.	Subjective tests	3
5.	Additional tests	4
6.	Other uses of the combiner	4
7.	Conclusions	5
8.	Acknowledgements	5
9.	References	5

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1. Introduction

It has been known for many years that, when receiving h.f. double-sideband transmissions using conventional receivers, there is a considerable advantage if diversity reception is employed. However, using modern independent-sideband, single-channel receivers with synchronous detectors, in place of receivers with envelope-detectors, the advantages of conventional diversity reception become marginal. In particular, there remains a considerable amount of distortion, especially under frequency-selective fading conditions. Such conditions occur when the signal arrives at the receiver by two or more paths with differing propagation delays; typical delay-differences are of the order of 1 ms. Path differences between the components of the received signal give rise to a number of minima in the modulation-frequency response, producing a 'comb-filter' effect. As small changes in path-length occur, the frequencies of the nulls or 'teeth' of the comb, will in general, move steadily through the r.f. band and, hence, through the audio band after demodulation. The signals picked up by two well-spaced aerials will both comprise the same set of differently delayed components, but the phases between these components will be different at the two aerials. Thus, although still showing a comb-filter effect, the two demodulated signals are likely to have frequencies for the teeth of the comb that are different or shifted relative to each other. Intuitively, it is expected that it should be possible to combine two such signals so as to obtain a better overall response. To find a satisfactory way of combining the signals, however, is not simple.

Addition of the two demodulated audio signals results in a new comb-filter structure of similar form and, on average, does not give any improvement to the output signal. If the phase of one of the audio signals is changed before addition, then the resulting comb-filter structure will be shifted in frequency. If, however, the phase of one audio signal is continuously steered by suitable means, so as to co-phase the principal audio-signal components of the two signals, then the comb-filter minima will affect less-important parts of the audio spectrum and the overall quality of the combined output will be improved.[1] The correlation audio combiner provides an automatic method of achieving and maintaining this kind of co-phasing action.

Following the development of the original experimental equipment,[1] a new prototype combiner has been constructed with a better optimisation of the phasing time-constants and additional circuits that further reduce the variations of output signal level. The following Sections describe the improved prototype combiner together with the results of some subjective tests that were conducted to assess its performance.

2. Correlation audio combiner

In Section 1 it was stated that, by varying the phase of one of the audio signals before addition, it was possible to vary the frequencies at which nulls occurred in the spectrum of the combined signal. If this phase difference between the two channels is continuously adjusted, so that the principal audio-frequency components are kept in phase, then, for many programmes, the nulls will fall at frequencies where there is little energy. In these circumstances, any frequency-selective distortion will be less noticeable than if the null had occurred at or near the frequency of the major component.

This method of combination forms the basis for the correlation audio combiner.*

The correlation audio combiner has two main sections. One is the correlation and combining section; the other is a section which both selects whichever of the two input signals has the larger long-term mean level (termed the master signal) and provides overall gain compensation, depending upon the ratio of the signal levels in the two channels. The former section is similar to the original experimental combiner[1] and, for simplicity, will be described first.

2.1. Correlation and combining

This section of the correlation audio combiner is shown on the right-hand side of Fig. 1. Its inputs comprise the two audio signals A′ and B′, where A′ is the master signal. These two audio signals are applied to a pair of quadrature phase-difference networks[2] (1a, 1b). One of these networks has a pair of outputs with a quadrature phase relationship between them which is maintained throughout the audio-frequency band of 80 Hz to 6 kHz. Both of these signal outputs suffer an unavoidable additional phase-shift, which increases with frequency, and the purpose of the complementary network in the A′ channel is to compensate for this phase-shift by introducing into the A′ signal path the same phase-shift as that imposed on the nominally zero-phase output in the B′ path.

The outputs of the quadrature phase-difference networks are used for two main purposes. One is to provide the audio programme signal feed to the linear multipliers (2a, 2b) and summing unit (3), whilst the other is to feed the 300 Hz to 2 kHz bandpass filters (4a, 4b, 4c) followed by clippers (5a, 5b, 5c). The clippers act as non-linear amplitude limiters and each produces a substantially square-

* PHILLIPS, G.J., SUSANS, D.E. Improvements in the combining of nominally similar electrical audio signals. British Patent Application No. 46928/75.

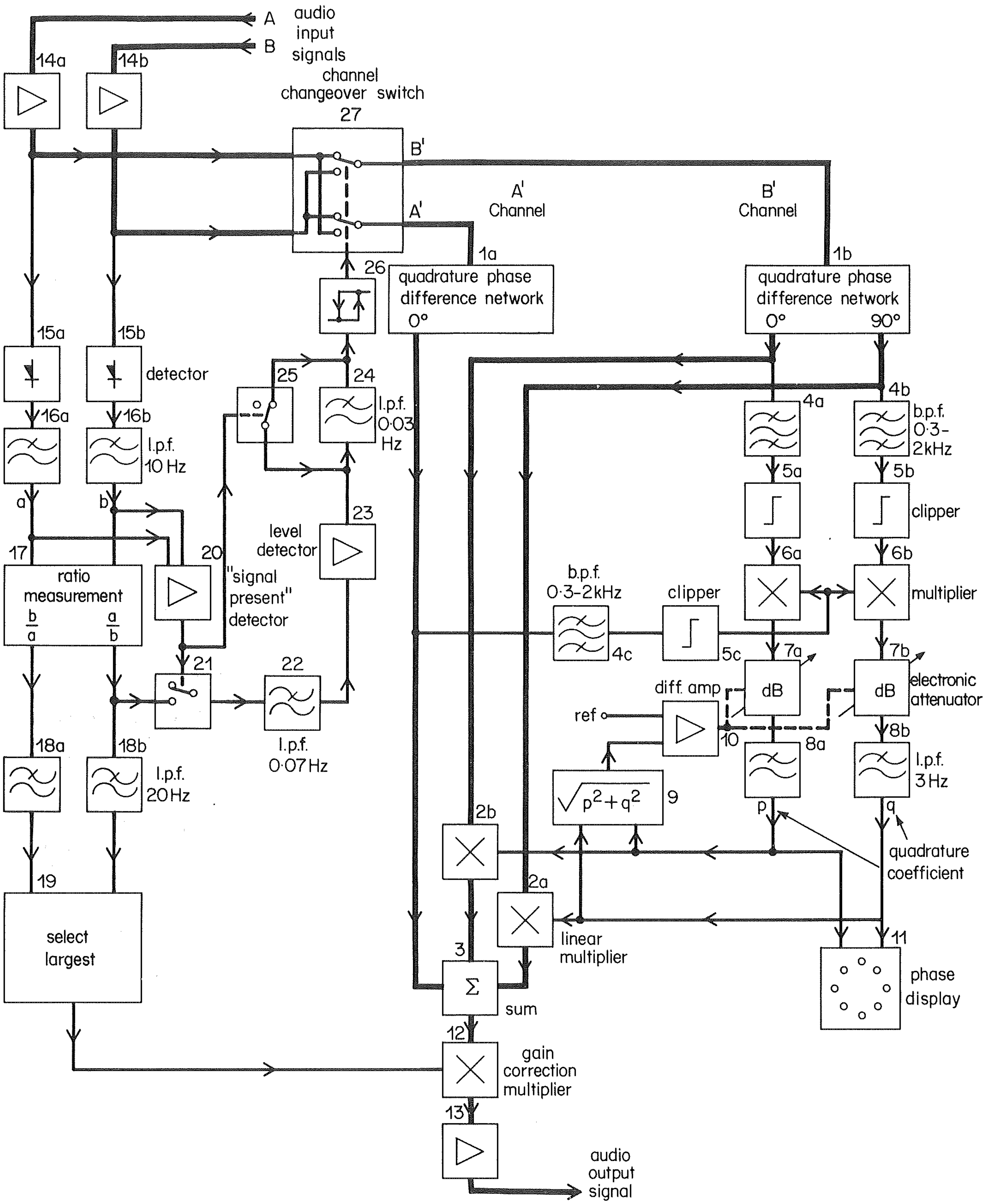

Fig. 1 - Correlation audio combiner

wave output signal over a wide range of input signal levels. The outputs of the clippers 5a and 5b in the B′ channel are multiplied by the output signal of the A′ channel clipper 5c, in the multipliers 6a and 6b, and the respective product-signals are passed through 3 Hz low-pass filters 8a, 8b. (We ignore for the moment the electronic attenuators 7a and 7b.) The outputs 'p' and 'q' coefficient for the two audio input signals A′ and B′. They are used by the linear multipliers (2a, 2b) to control the levels of the two quadrature components of the B′-channel signal that are added to the A′-channel signal in the final combiner (3). The components p and q of the correlation coefficient are also used to drive a ring of eight lamps (11) to provide a visual indication of the instantaneous phase relationship between the input signals A′ and B′.

Under severe multipath conditions, the modulus of the complex correlation coefficient may become very small. This would result in an incorrect contribution from the B′ channel. Accordingly, the modulus of the correlation coefficient is computed from p and q (9) and compared with a reference signal in the differential amplifier (10). The error signal output is used to control electronic attenuators (7a, 7b) to form an automatic level-control loop which maintains the modulus of the correlation coefficient at an approximately constant value.

In the event of a signal fade in one channel, or a pause in the programme, the correlation data could be lost and the circuit could restart with an arbitrary combining phase. A 'cross-feed' unit (not shown in Fig. 1) is therefore incorporated which, during a fade, maintains the values of p and q close to the mean values which they held during the previous 30 seconds.

2.2. Gain-compensation

When fading is present, the basic action of the correlation combiner has the beneficial effect of reducing the range of variations of the output signal level. Further reduction in the range of output signal levels is possible by use of the following gain-compensation technique.

The two co-phased audio signals add arithmetically and this, without gain-compensation, could result in an increase in output level of up to 6 dB above that of the larger of the two input signals, depending on the amplitude of the other signal. To reduce this unwanted variation in output level, gain-compensation is provided. In order to be independent of programme content, the gain-compensation is controlled by the ratio of the amplitudes of the two input audio signals (A and B in Fig. 1), rather than by their absolute amplitudes. In practice, a full 6 dB gain reduction for equal signals should not be applied. This is because, at times during the fading cycle when the two signals are equal, it is unlikely that both of them will achieve their peak values simultaneously. The optimal correction is between 4·5 dB and 5 dB for inputs of equal level.

The ratio of the amplitudes of the two input signals is measured as follows. The two audio signals (A and B) are fed to signal-level detectors, consisting of full-wave rectifiers (15a, 15b) followed by 10 Hz low-pass filters (16a, 16b). The outputs from these two detectors are then used to control the durations of the two periods of an astable multivibrator operating as a ratio-measuring circuit (17). The duration of each period of the multivibrator is inversely proportional to the output from the associated detector. Thus the mark-to-space ratio of the multivibrator is a non-linear function of the ratio of the two input signal levels. Two complementary multivibrator output signals are formed, each of which thereby describes the ratio and inverse ratio, respectively, of the input audio signal levels. Thus low-pass filters (18a, 18b), fed from these two multivibrator outputs, deliver output voltages that are dependent on the ratio and the inverse ratio of the amplitudes of the two audio input signals. A simple circuit then selects the larger of these two outputs for use as a control voltage for the gain-correction multiplier (12). The non-linear relationship that results from the ratio measurement is a convenient match to that required for gain-correction.

The gain-correction multiplier (12) is followed by a standard output amplifier (13).

2.3. Master signal selection

Additional outputs are taken from the 10 Hz low-pass filters (16a, 16b) to drive a 'signal present' detector (20). When either signal is present, a switch (21) is closed which connects one output from the ratio measurement unti to a low-pass filter (22) which has a 15-second integration time. A simple level-detector (23) is fed from this filter and is used to operate the channel changeover switch (27), according to the integrated ratio measurement so obtained. The operation of the channel changeover switch is such as to select the signal A′ from whichever of the two input signals A or B has the greater long-term mean value. Operation of this switch is delayed for a half-minute by a low-pass filter (24) or until an earlier break in the programme occurs; this reduces the possibility of a click being heard during the changeover.

3. Use of receiver A.G.C.

Operational experience with a correlation audio combiner has shown that it is not practicable to link the a.g.c. circuits on the two channels of diversity receivers, as was originally proposed.[1] This is because of differences between the receiver channel-gains and differences between the aerials, such that the required gain equality could not be maintained. As a result, it is now recommended that the two diversity receiver channels should be operated completely independently.

4. Subjective tests

Subjective tests were performed using dual-track audio tape recordings. Some of these recordings were made at Caversham of transmissions received from Greenville U.S.A. but the majority were made at the BBC Far Eastern Relay Station at Singapore of transmissions from the UK.

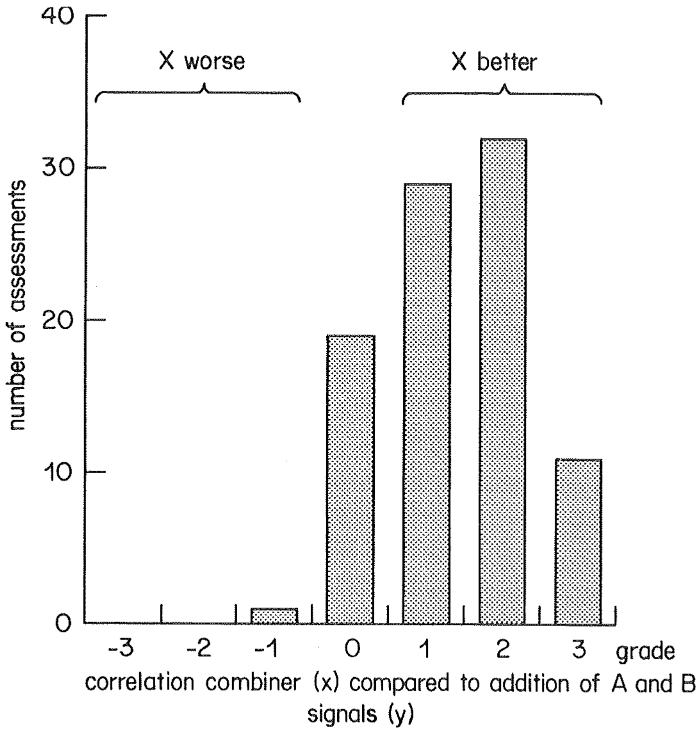

Fig. 2 - Comparison between the performance of the combiner and simple addition of channels A and B

Grade	
3	X much better than Y
2	X better than Y
1	X slightly better than Y
0	no difference
−1	X slightly worse than Y
−2	X worse than Y
−3	X much worse than Y

Plessey PRD200A and PRD200 dual-diversity s.s.b. receivers were used. Unfortunately, the receiver used at Singapore was not ideal, in that it employed a common a.g.c. circuit for both channels; for the reasons given in Section 3, it was known that this might reduce the performance of the combiner, but it was the only receiver available.

The two tracks of the tape recorder carried the individual audio signal outputs from the two channels of the diversity receiver, thus permitting various methods of audio combining to be compared. In the tests described here, comparisons were made between the output of the correlation audio combiner and the signal obtained by simple addition of the two audio signals.

The programme material used for the tests consisted of short excerpts from a wide selection of speech and music programmes. The same excerpts were combined in each of the two ways and the order of presentation of the various conditions to the listeners was changed in a random manner. Twelve listeners were asked to compare the results using the CCIR comparison scale[3] shown in Fig. 2.

As can be seen from Fig. 2, the results showed an average improvement in favour of the combiner of about 1½ subjective grades. A more detailed examination of the results and subsequent questioning of the listeners showed that the combiner gave little improvement (but no degradation) for pop music and full orchestra, in which there is rarely one predominant frequency. For speech and single singers, however, the combiner gave a marked improvement. Listeners with considerable experience of h.f. rebroadcast circuits could identify the use of the combiner within a few seconds and commented on the improvement in quality and reduction in listener fatigue. Some listeners commented on the reduced fading range; this is typically about 3 dB, relative to that of the combiner without gain-compensation.

5. Additional tests

In addition to the formal subjective tests, other tests of an informal nature have been undertaken. These tests, mainly carried out by experienced h.f. listeners, have all confirmed the findings of the formal tests.

In an experiment to assess the degradation in quality of a programme after passing through the combiner, high-quality recordings were used for the two audio input signals. When the two inputs were identical, no degradation could be heard. If one input was delayed by 1 ms and continuously rotated in phase (7 radians/sec), the combiner output sounded as if it had suffered some loss of bass and high frequencies, but was otherwise of good quality. When the same two inputs were added directly, the distortion was very serious. This test corresponds to a limiting case of selective fading.

In an 'off-air' test, a fast (1 Hz) flutter fade was present which, on either a single-channel receiver or in the (A + B) mode, made it difficult to identify the language being spoken. When the combiner was used, the flutter fade was virtually eliminated and apart from a rather high noise-level, the programme was suitable for rebroadcast purposes.

Some recent field trials of the combiner carrier out at the British Eastern Mediterranean Relay Station showed it to provide appreciable improvements in quality for both speech and music.

6. Other uses of the combiner

It has been suggested that the correlation audio combiner could be used for other types of signals in the h.f. band.

One such application would be for telephony signals. Here, the lowest frequency required is about 300 Hz and a slightly higher level of distortion is permissible. As a result, the low-pass filters (8a and 8b in Fig. 1) could be increased in bandwidth by about 5 : 1, giving a better response under conditions of fast flutter fading. For use with frequency-scrambled telephony signals, it would also be advisable to increase the bandwidths of the filter 4a, 4b and 4c so as to pass the full 3·4 kHz range of telephony signal frequencies.

The correlation audio combiner also appears to be very suitable for f.s.k. telegraphy signals. Here, the band-

widths of the low-pass filters 8a and 8b could be increased to give a faster response than would be tolerable for telephony or programme signals.

7. Conclusions

An improved correlation audio combiner for use with h.f. dual-diversity receivers has been described, together with subjective tests on this equipment. In conditions of frequency-selective fading, these tests have shown improvements in programme quality of about 1½ subjective grades. The greatest improvements are for speech and the least improvements for 'pop' music and orchestral items. The improvements give both a reduction of frequency-selective distortion and a reduced fading range. Other applications of the combiner for telephony and telegraphy are also possible.

8. Acknowledgements

Thanks are due to BBC External Broadcasting Department and the BBC Far Eastern Relay Station for the special recordings used for the subjective tests described in this Report.

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Printed by BBC RESEARCH DEPARTMENT, Kingswood Warren, Tadworth, Surrey, KT20 6NP